MITEL – SIP CoE

Technical
Configuration Notes

Configure MCD 6.0 SP3 for use with BT GS

SIP COE 11-4940-00172

MITEL

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NOTICE

Mitel Technical Configuration Notes:

Configure the Mitel MCD 6.0 SP3 for use with BT GS
April 2014 11-4940-00172_2

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 3300 MCD to connect to BT GS. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

Interop History

Version	Date	Reason
1	June 2011	Initial Interop with Mitel 3300 MCD 4.2 SP2 and BT GS
2	April 2014	Interop with Mitel 3300 MCD 6.0 SP3 and BT GS

Interop Status

The Interop of BT GS has been given a Certification status. This service provider or trunking device will be included in the SIP CoE Reference Guide. The status BT GS achieved is:

COMPATIBLE	The most common certification which means BT GS has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between BT GS and the 3300 MCD.

Manufacturer	Variant	Software Version
Mitel	3300 MCD 6.0 SP3 – Mxe Platform	12.0.3.15
Mitel	MBG - Gateway	8.0.26.0
Mitel	MBG - Teleworker	8.0.17.0
Mitel	Nupoint Voicemail	14.1
Mitel	5330 SIP Sets	SIP (05.02.03.01)
Mitel	5330/5340 IP Sets	Minet (05.02.03.01)
PSTN Gateway	CISCO PGW	
SBC	ACME 4250	ACME Net-Net 4250 Firmware SC6.1.0 MR-11 patch 1 (Build 1036

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Trunk Side Interoperability Test Pans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through BT GS and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	☑
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	☑
Packetization	Forcing the 3300 MCD to stream RTP packets through its E2T card at different intervals, from 10ms to 60ms	⚠
Personal Ring Groups	Receiving calls through BT GS and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑
Teleworker	Making and receiving a call through BT GS and their PSTN gateway to and from Teleworker extensions.	☑
Video	Making and receiving a call through BT GS with video capable devices.	✗
Fax	T.38 and G711Fax Calls	☑

☑ - No issues found ✗ - Issues found, cannot recommend to use ⚠ - Issues found

Device Limitations and Known Issues

This is a list of problems or not supported features when BT GS is connected to the Mitel MCD 6.0 SP3.

Feature	Problem Description
Video	BT GS does not support video calls over SIP Trunks. **Recommendation:** Consult BT GS for future deployment of SIP video calls.
Packetization	3300 MCD was tested with stream RTP packets through its E2T card at different intervals of 20ms to 30ms (BT recommendation) instead of the Mitel recommended range of 10ms to 60ms. **Recommendation:** Use BT GS 20ms or 30ms recommended setting.
Loopback Testing and RFC 3261 Call-ID 8.1.1.4	During standard Mitel SIP trunking test setup we had an issue with all DID loopback calls, whereby the inbound INVITE back to the MBG from BT re-uses the same call-id and the MBG responded back with a 491 request pending. **Recommendation:** Mitel strongly suggests that re-using a call ID violates a MUST and should be fixed. As a result no loopback calls were tested and all test calls were therefore limited to either originate/terminate from a Mitel (SIP, IP and Analog) phone set or FAX to/from a respective private BT GS PSTN, IVR or FAX extension.

Network Topology

This diagram shows how the testing network is configured for reference.

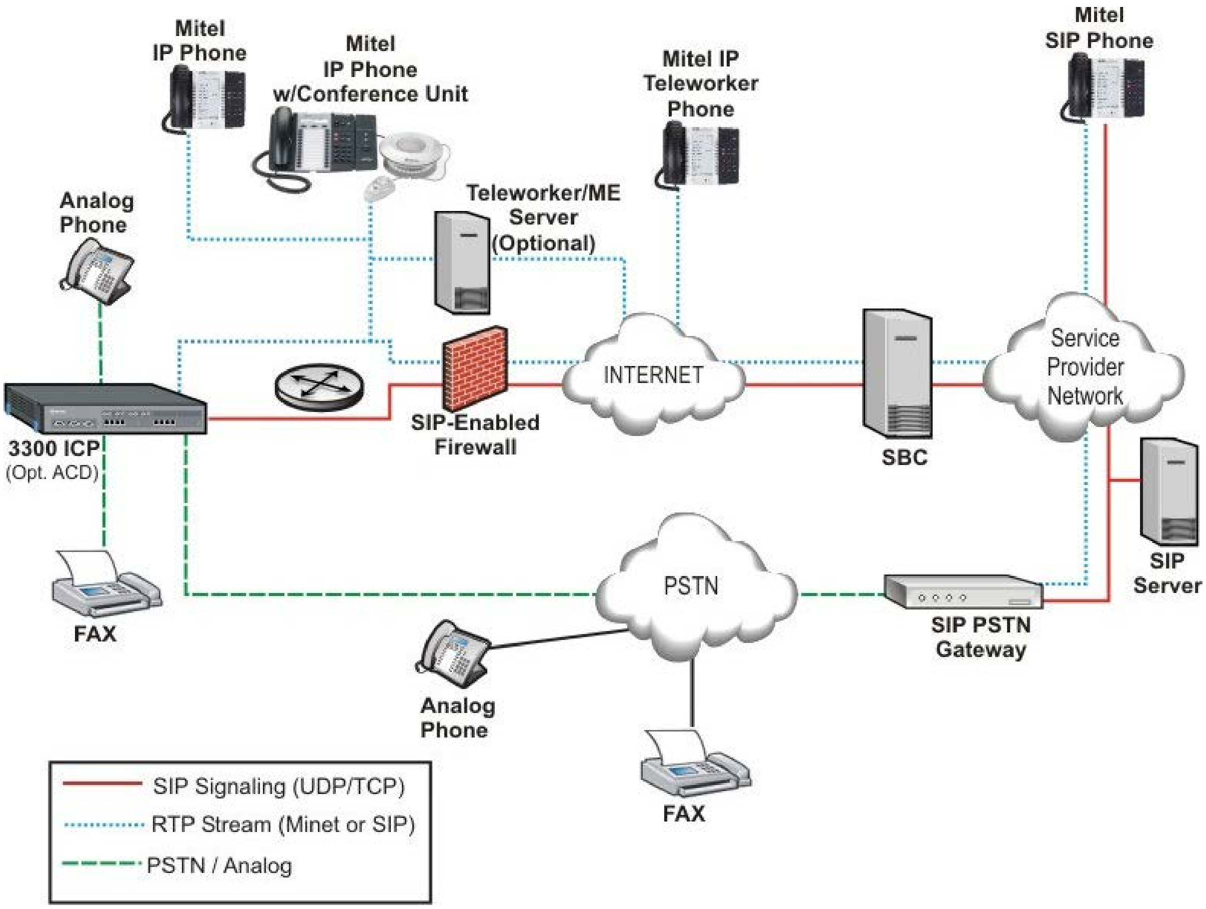


Figure 1 – Network Topology

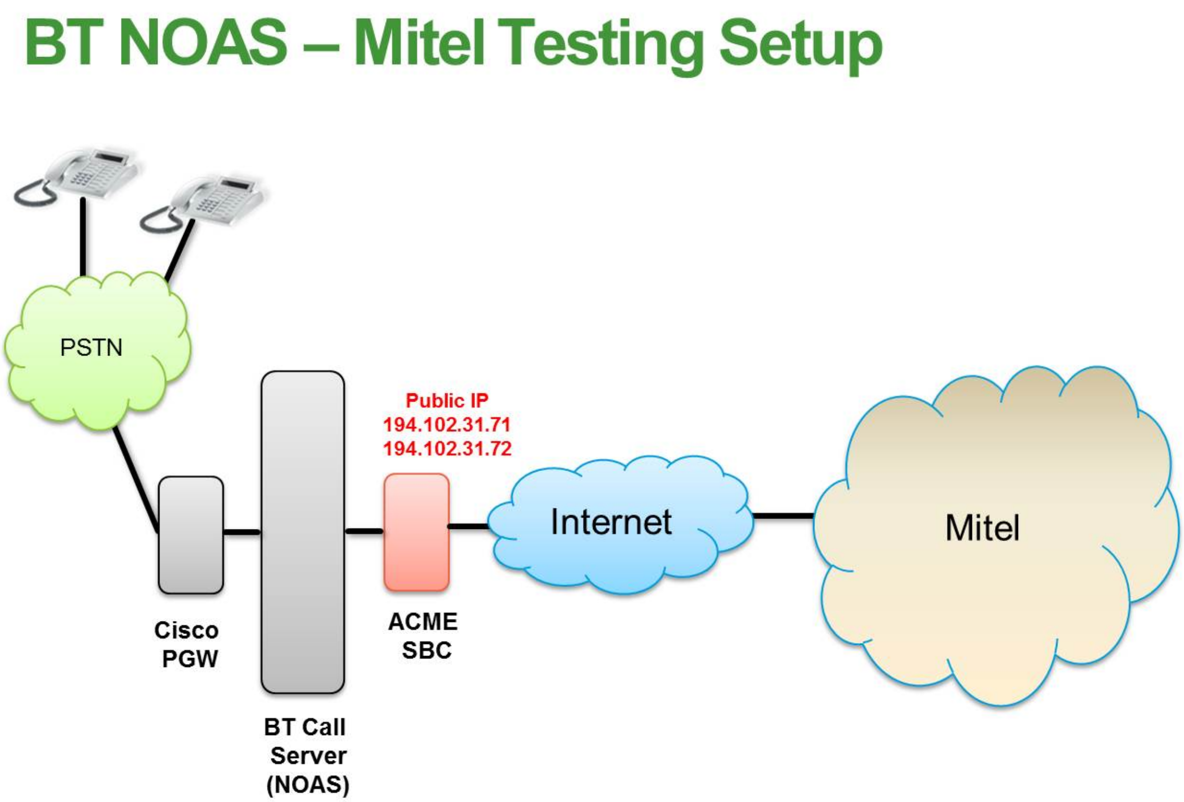
Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how BT GS 3300 programming was configured in our test environment.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

BT GS Configuration Notes

The BT GS testing environment was an internal/private Sandbox setup with no connection to outside public network other than the Mitel MBG via redundant BT SBC.



SIP Service Provider Server IP address	SBC IPs: 194.102.31.71 and 194.102.31.72
Media Server	N/A
Registration and Authentication	N/A
Pilot Number	N/A
Username/Password	05511500200
DIDs	05511500200-05511500202
PSTN	01912500753-01912500754
FAX	01912226651
IVR	08207232007
Preferred Codec	G711A,G711u,G729
SIP port	5060
Transport Type	UDP
Session Timer	Requests will be ignored

3300 MCD Configuration Notes

The following steps show how to program a 3300 MCD to interconnect with BT GS.

Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the 3300 Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

Assumptions for the 3300 MCD Programming

- The SIP signaling connection uses UDP on Port 5060

Licensing and Option Selection – SIP Licensing

Ensure that the 3300 MCD is equipped with enough SIP trunking licenses for the connection to BT GS. This can be verified within the License and Option Selection form.

Enter the total number of licenses in the SIP Trunk Licences field. This is the maximum number of SIP trunk sessions that can be configured in the 3300 to be used with all service providers, applications and SIP trunking devices.

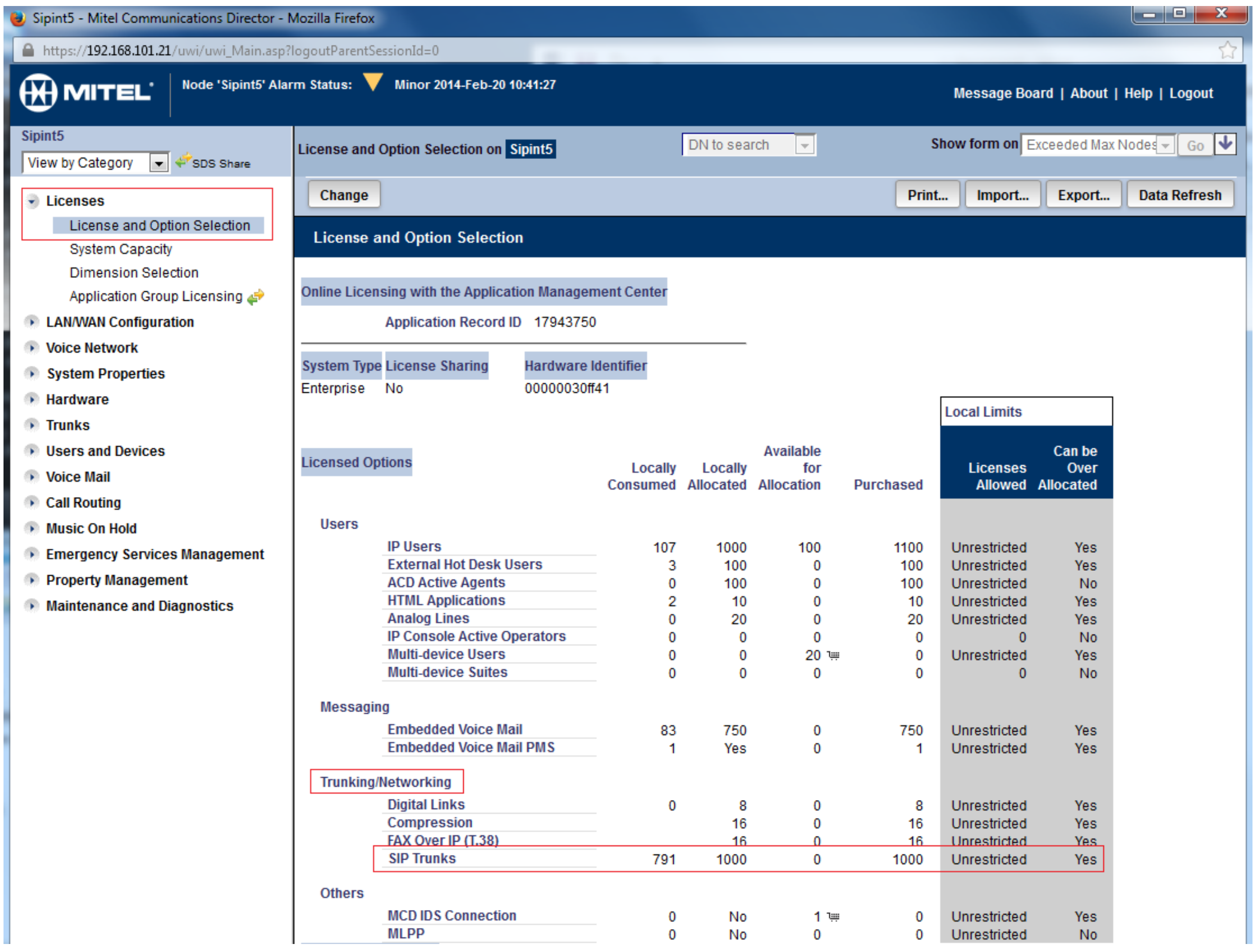


Figure 2 – License and Option Selection

Class of Service Assignment

The Class of Service Options Assignment form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Service Assignment form for SIP trunks.

Many different options may be required for your site deployment, but ensure that “Public Network Access via DPNSS” Class of Service Option is configured for all devices that make outgoing calls through the SIP trunks in the 3300.

- Public Network Access via DPNSS set to **Yes**
- Campon Tone Security/FAX Machine set to **Yes**
- Busy Override Security set to **Yes**
- Clear Auto Campon Timer to enable 486 BUSY testing

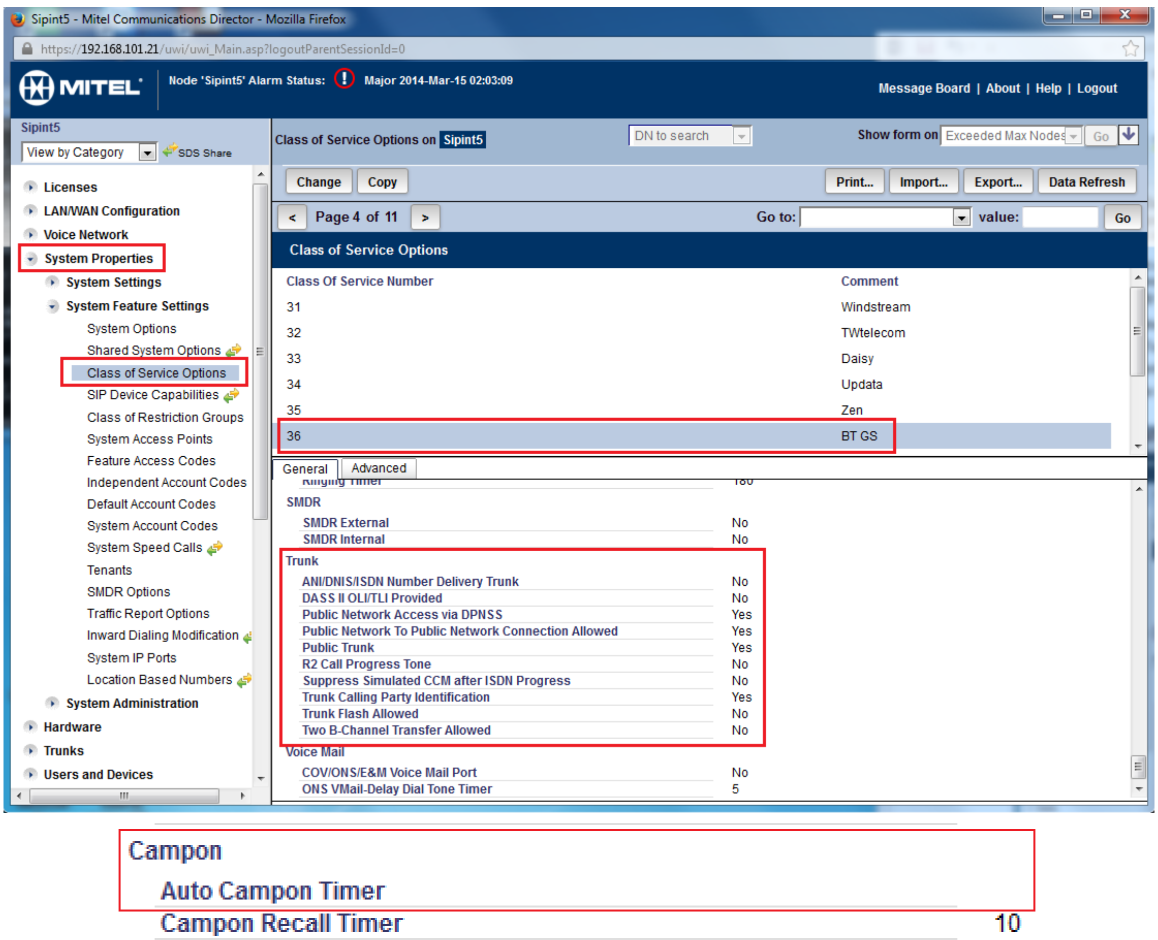


Figure 3 – Class of Service

Network Element Assignment

Create a network element for BT GS. In this example, the softswitch is reachable by an IP Address and is defined as “BT GS” in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy and Registrar are provided by your service provider.**

If your service provider trusts your network connection by asking for your gateway external IP address, then programming the IP address for the SIP Peer, Outbound Proxy and Registrar is not required for SIP trunk integration. This will need to be verified with your service provider. Set the transport to UDP and port to 5060.

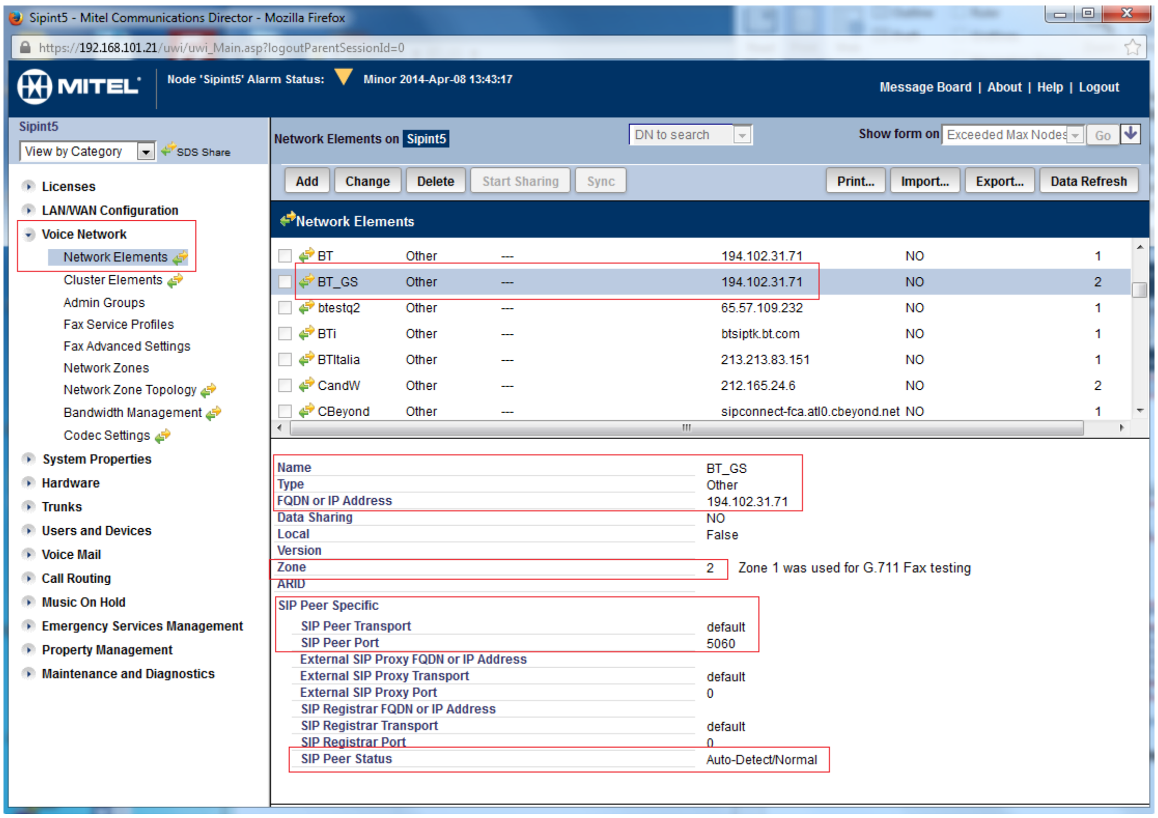


Figure 4 – Network Element Assignment

Network Element Assignment (Proxy)

In addition, depending on your configuration, a Proxy may need to be configured to route SIP data to the service provider. If you have a Proxy server installed in your network, the 3300 ICP will require knowledge of this by programming the Proxy as a network element then referencing this proxy in the SIP Peer Profile form (later in this document).

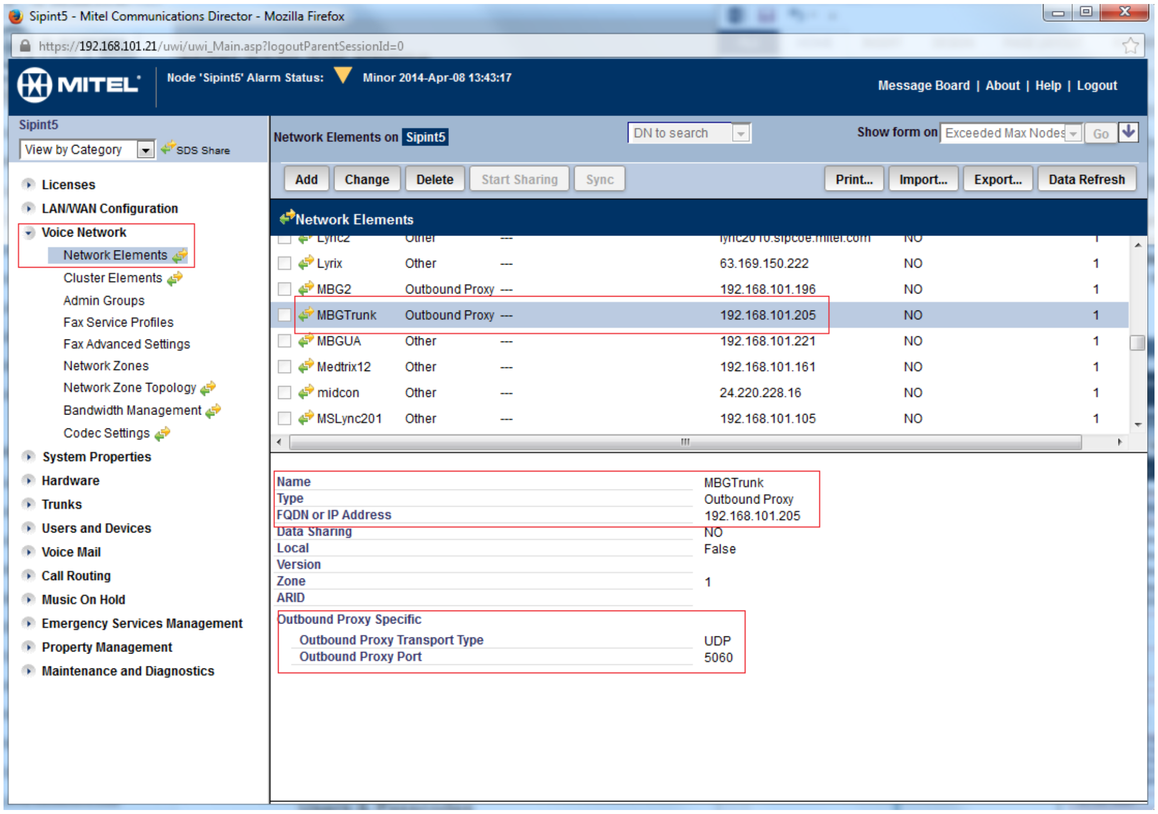


Figure 5 – Network Element (Proxy)

Trunk Service Assignment

This is configured in the Trunk Service Assignment form. In this example the Trunk Service Assignment is defined for Trunk Service Number 25 which will be used to direct incoming calls to an answer point in the 3300.

Program the Non-dial In or Dial In Trunks (DID) according to the site requirements and what type of service was ordered from your service provider.

The example below shows configuration for incoming DID calls. The 3300 will absorb the first 7 digits of the DID number from BT GS leaving 5 digits for the 3300 to translate and ring a 4 digit extension. For example, BT GS delivers +44-551-150-0200 through the SIP trunk to the 3300. The 3300 will absorb the first 7 digits (4455115) leaving the 3300 to ring extension 00200. Extension 00200 must be programmed as a valid 4 digit dialable number in the 3300 via system speed call. Please refer to the 3300 System Administration documentation for further programming information.

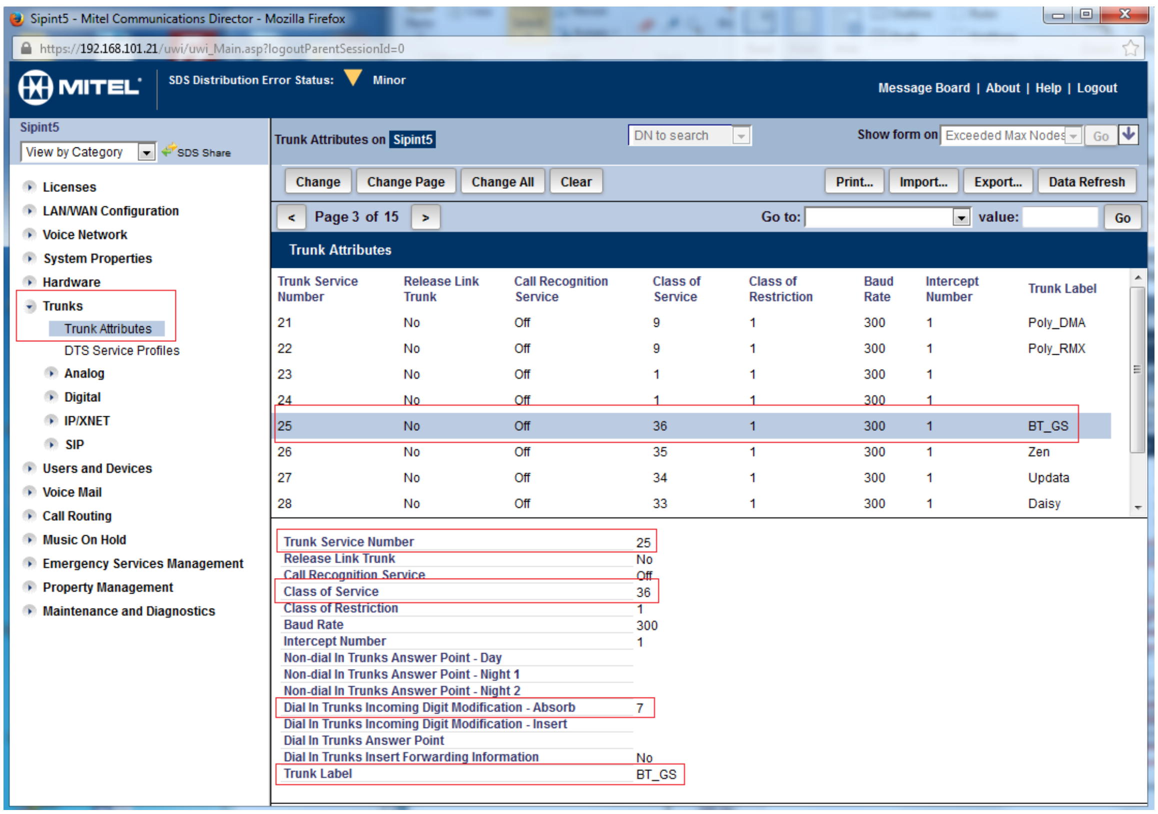


Figure 6 – Trunk Service Assignment

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base 3300 MCD Platform. The SIP Peer Profile should be configured with the following options:

Network Element: The selected SIP Peer Profile needs to be associated with previously created “BT GS” Network Element.

Registration User Name: BT GS does require the use of a Registration User Name.

Address Type: Select IP address.

Default CPN: The default CPN “05511500200” is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided** by BT GS. Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the DID ranges for CPN Substitution form (see DID Ranges for CPN Substitution). Then select the appropriate numbers in the Outgoing DID Ranges in this form (SIP Peer Profile).

Trunk Service Assignment: Enter the trunk service assignment previously configured.

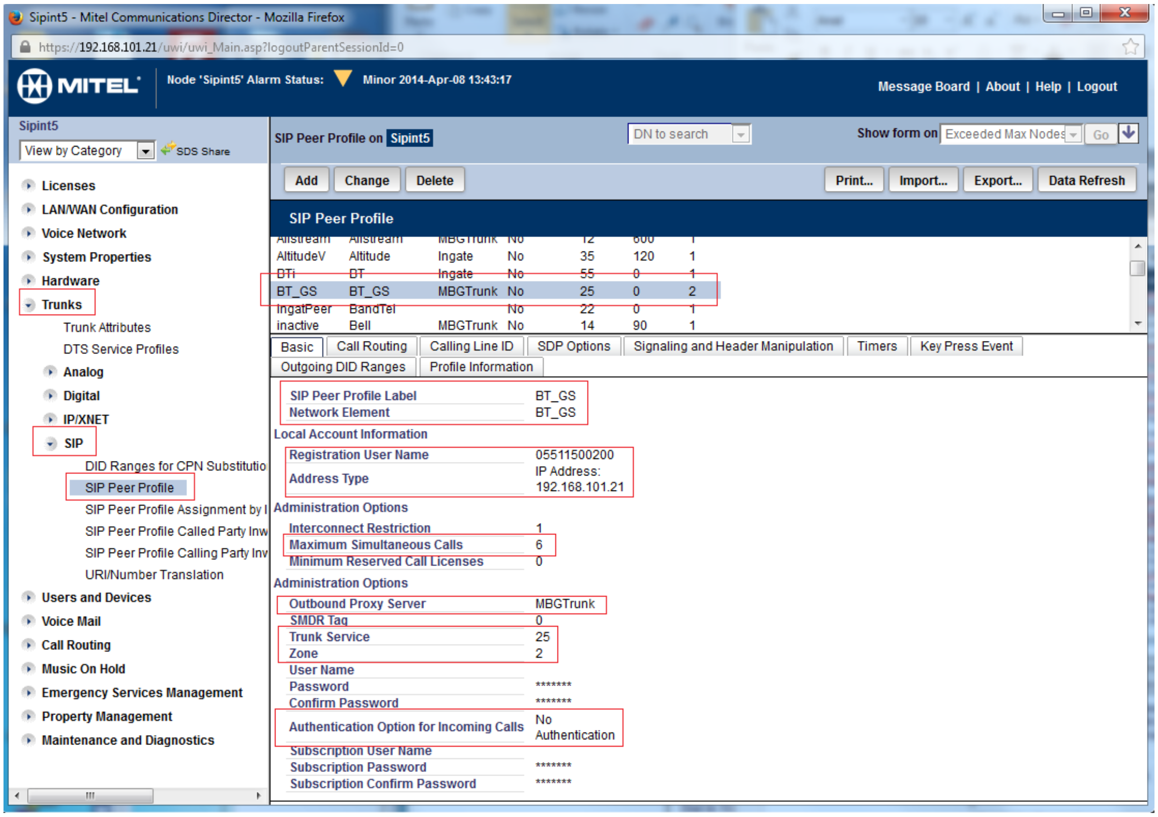
SMDR: If Call Detail Records are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank).

Maximum Simultaneous Calls: This entry should be configured to maximum number of SIP trunks provided by BT GS.

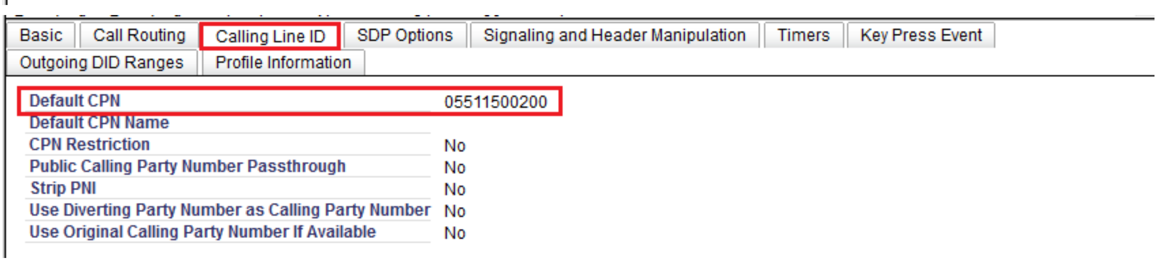
NOTE-1: Ensure the remaining SIP Peer profile policy options are similar the screen capture below.

NOTE-2: SDP options had Updates disabled (not supported by BT) and Force SDP sent in the initial invite enabled.

NOTE-3: BT Session Timer not supported or used and set to 0.


Sipint5 - Mitel Communications Director - Mozilla Firefox
https://192.168.101.21/uwi/uwi_Main.asp?logoutParentSessionId=0
MITEL
Node 'Sipint5' Alarm Status:
Minor 2014-Apr-08 13:43:17
Message Board | About | Help | Logout
Sipint5
View by Category
SDS Share
Licenses
LAN/WAN Configuration
Voice Network
System Properties
Hardware
Trunks
Trunk Attributes
DTS Service Profiles
Analog
Digital
IP/XNET
SIP
DID Ranges for CPN Substitutio
SIP Peer Profile
SIP Peer Profile Assignment by I
SIP Peer Profile Called Party Inw
SIP Peer Profile Calling Party Inv
URI/Number Translation
Users and Devices
Voice Mail
Call Routing
Music On Hold
Emergency Services Management
Property Management
Maintenance and Diagnostics
SIP Peer Profile on Sipint5
DN to search
Show form on Exceeded Max Nodes Go
Add Change Delete
Print... Import... Export... Data Refresh
SIP Peer Profile
AltitudeV Altitude Ingate No 35 120 1
DTi DT Ingate No 55 0 1
BT_GS BT_GS MBGTrunk No 25 0 2
IngatPeer BandTel No 22 0 1
inactive Bell MBGTrunk No 14 90 1
Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event
Outgoing DID Ranges Profile Information
SIP Peer Profile Label BT_GS
Network Element BT_GS
Local Account Information
Registration User Name 05511500200
Address Type IP Address: 192.168.101.21
Administration Options
Interconnect Restriction 1
Maximum Simultaneous Calls 6
Minimum Reserved Call Licenses 0
Administration Options
Outbound Proxy Server MBGTrunk
SMDR Tag 0
Trunk Service 25
Zone 2
User Name
Password *******
Confirm Password *******
Authentication Option for Incoming Calls No Authentication
Subscription User Name
Subscription Password *******
Subscription Confirm Password *******


Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event
Outgoing DID Ranges Profile Information
Alternate Destination Domain Enabled No
Alternate Destination Domain FQDN or IP Address
Enable Special Re-invite Collision Handling No
Only Allow Outgoing Calls No
Private SIP Trunk No
Reject Incoming Anonymous Calls No
Route Call Using To Header No


Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event
Outgoing DID Ranges Profile Information
Default CPN 05511500200
Default CPN Name
CPN Restriction No
Public Calling Party Number Passthrough No
Strip PNI No
Use Diverting Party Number as Calling Party Number No
Use Original Calling Party Number If Available No

Basic | Call Routing | Calling Line ID | **SDP Options** | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Allow Peer To Use Multiple Active M-Lines	Yes
Allow Using UPDATE For Early Media Renegotiation	No
Avoid Signaling Hold to the Peer	Yes
Enable Mitel Proprietary SDP	No
Force sending SDP in initial Invite message	Yes
Force sending SDP in initial Invite - Early Answer	No
Ignore SDP in Unreliable Provisional Responses	No
Limit to one Offer/Answer per INVITE	Yes
NAT Keepalive	Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
Renegotiate SDP To Enforce Symmetric Codec	No
Repeat SDP Answer If Duplicate Offer Is Received	No
RTP Packetization Rate Override	No
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	No
Suppress Use of SDP Inactive Media Streams	No

Basic | Call Routing | Calling Line ID | SDP Options | **Signaling and Header Manipulation** | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Trunk Group Label	
Allow Display Update	No
Build Contact Using Request URI Address	No
De-register Using Contact Address not *	Yes
Disable Reliable Provisional Responses	No
Disable Use of User-Agent and Server Headers	No
E.164: Enable sending '+'	No
E.164: Add '+' if digit length > N digits	0
E.164: Do not add '+' to Emergency Called Party	No
E.164: Do not add '+' to Called Party	No
Force Max-Forward: 70 on Outgoing Calls	No
If TLS use 'sips:' Scheme	No
Ignore Incoming Loose Routing Indication	No
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Require Reliable Provisional Responses on Outgoing Calls	Yes
Suppress Redirection Headers	No
Use Fixed Retry Time for 491	No
Use Privacy: none	No
Use P-Asserted Identity Header	Yes
Use P-Asserted Identity for Billing	No
Use P-Preferred Identity Header	No
Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No
Use user=phone	No

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | **Timers** | Key Press Event | Outgoing DID Ranges | Profile Information

Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	0
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | **Key Press Event** | Outgoing DID Ranges | Profile Information

Allow Inc Subscriptions for Local Digit Monitoring	No
Allow Out Subscriptions for Remote Digit Monitoring	No
Force Out Subscriptions for Remote Digit Monitoring	No
Request Outbound Proxy to Handle Out Subscriptions	No
KPML Transport	default
KPML Port	0

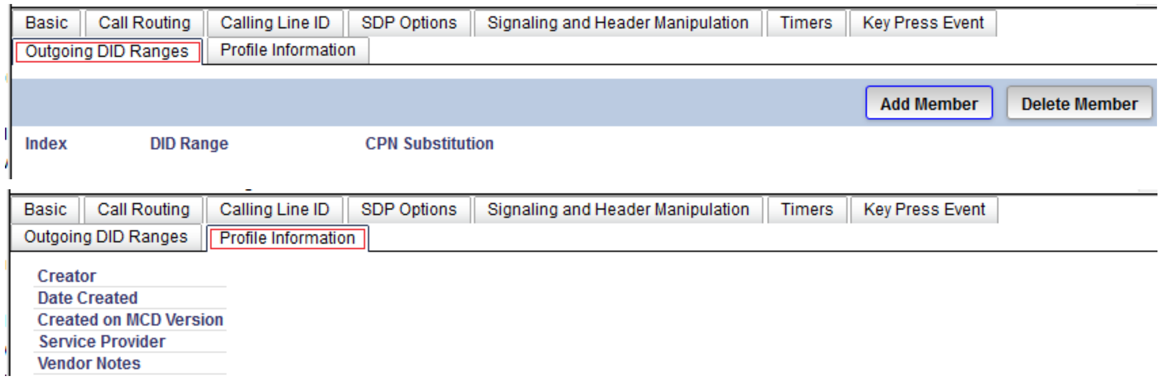


Figure 7 – SIP Peer Profile Assignment

SIP Peer Profile Assignment by Incoming DID

This form is used to assign incoming digits from BT GS. DID range numbers assigned by BT GS and are associated to a particular SIP Peer.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges or single numbers (for example, "01661520417,01282505152"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 character maximum, you can create a new entry for the same profile.

Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier", and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

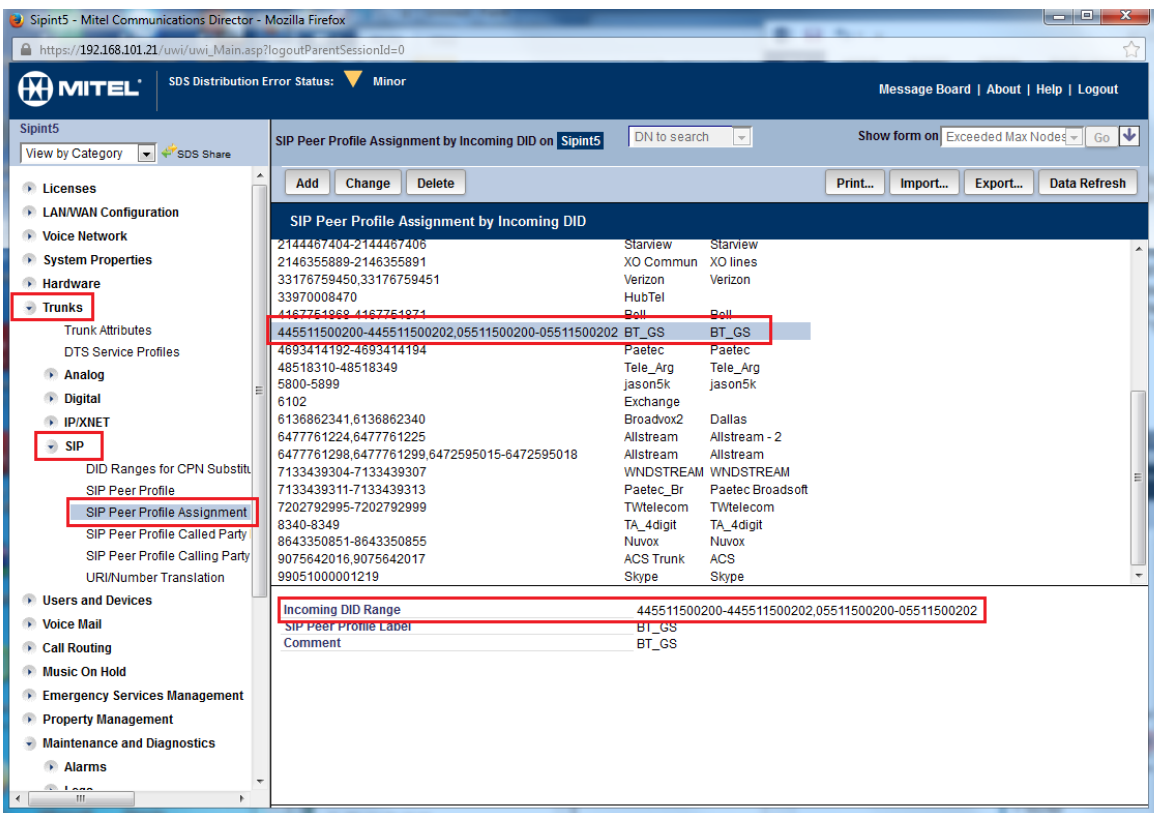


Figure 8 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Number

Ensure that Digit Modification for outgoing calls on the SIP trunk to BT GS absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 3 digits for both prefixes used: 924 prefix to dial out followed by 11 digits and 925 prefix followed by 13 digits.

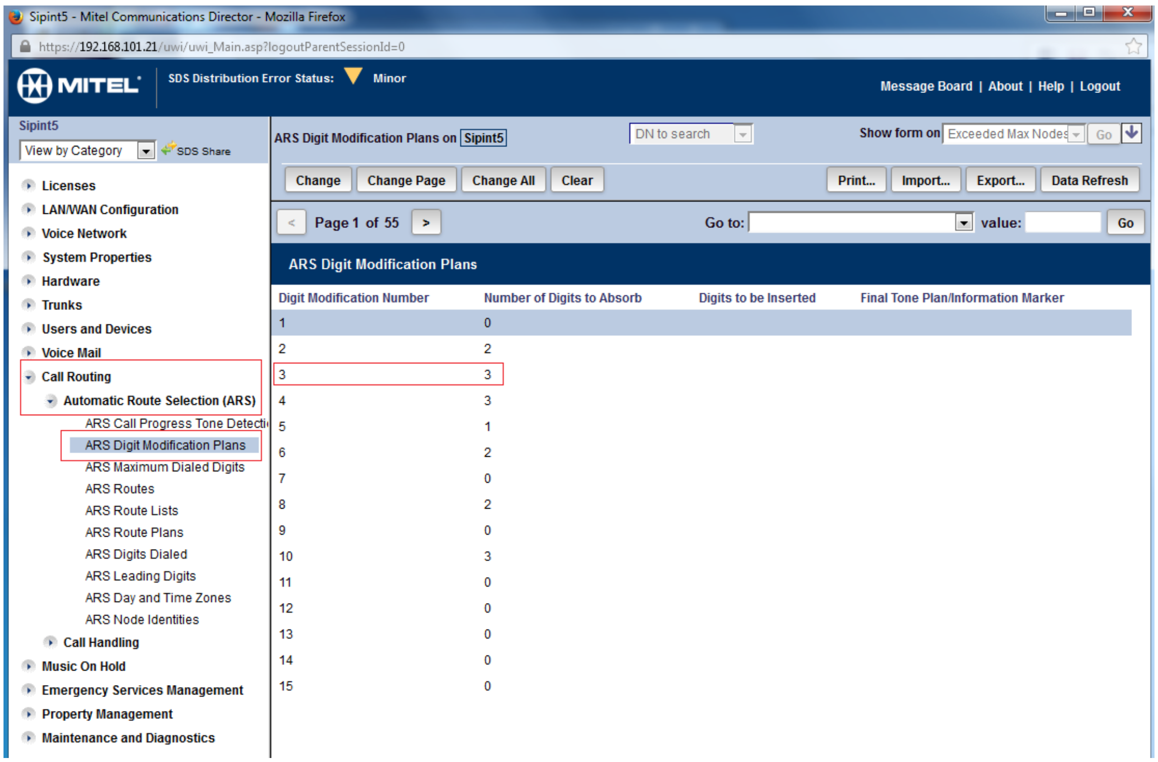


Figure 9 – Digit Modification Assignment

ARS Routes Assignment

Create a route for SIP Trunks connecting a trunk to BT GS. In this example, the SIP trunk is assigned to Route Number 31. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

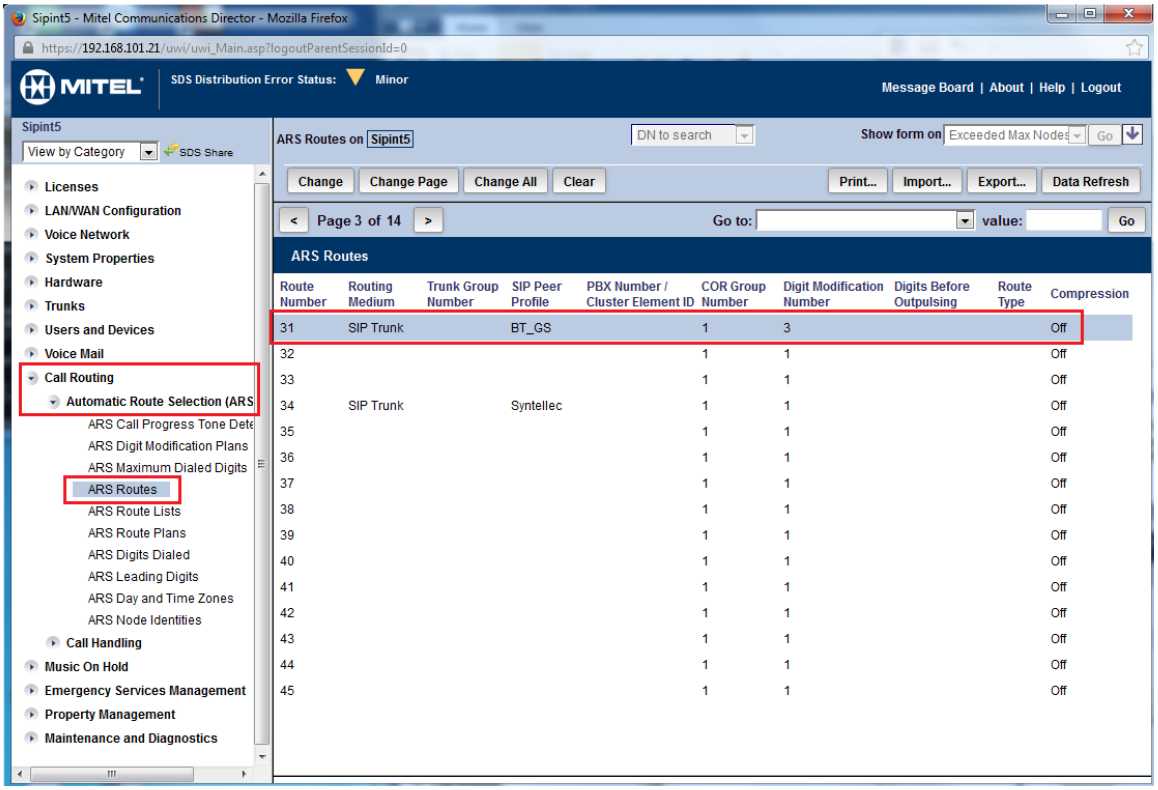


Figure 10 – SIP Trunk Route Assignment

ARS Digits Dialed Assignment

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 924 or 925, the call will be routed to BT GS (ie. Route 31).

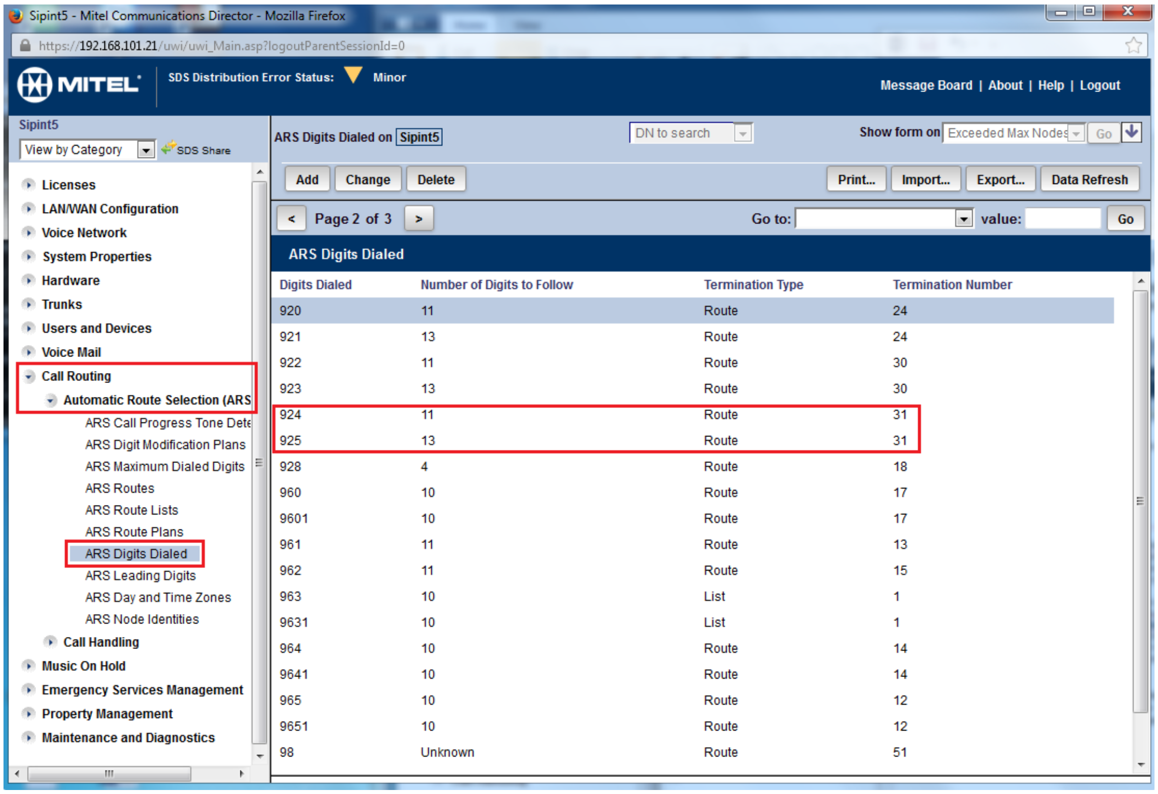


Figure 11 – ARS Digit Dialed Assignment

Fax Configuration

BT GS uses the inter-zone FAX profile. This form allows you to define the settings for FAX communication over the IP network. You can modify the default settings for the:

Inter-zone FAX profile: defines the FAX settings between different zones in the network. There is only one Inter-zone FAX profile; it applies to all inter-zone FAX communication. It defaults to V.29, 7200bps. It defines the settings for FAX Relay (T.38) FAX communication.

Intra-zone FAX profile: defines the FAX settings within each zone in the network.

- Profile 1 defines the settings for G.711 pass through communication.
- Profile 2 to 64 define the settings for FAX Relay (T.38) FAX communication.
- All zones default to G.711 pass through communication (Profile 1).

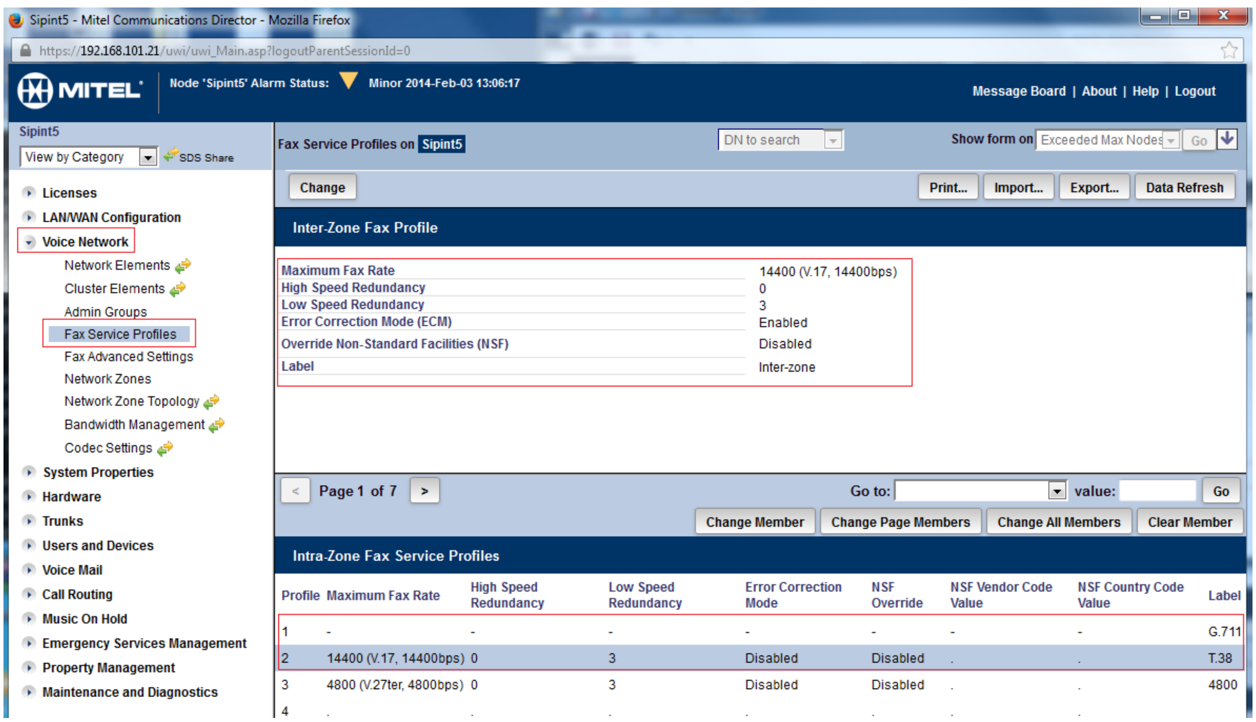


Figure 12 – Fax Service Profiles

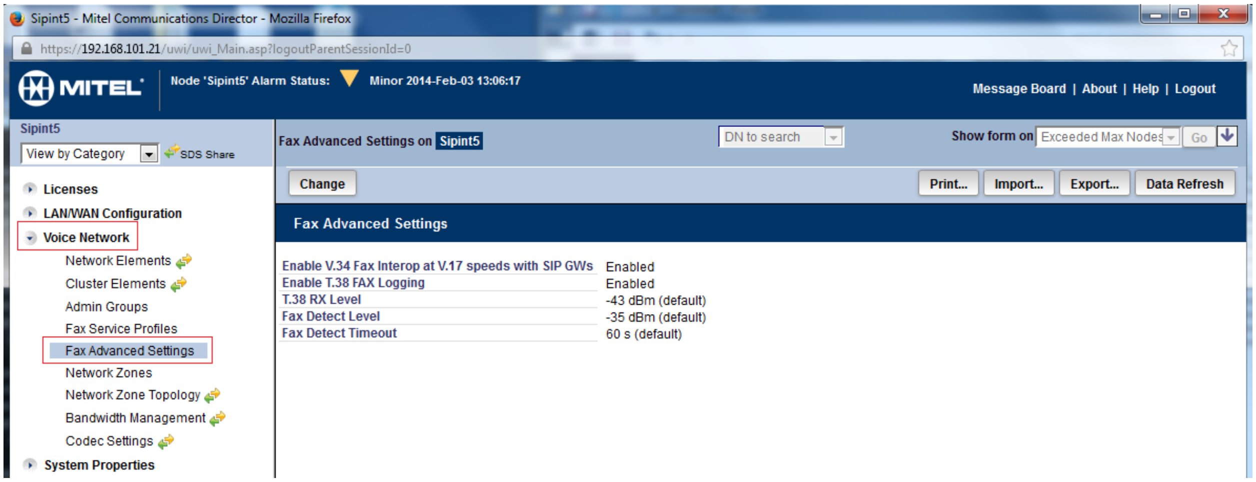


Figure 13 – Fax Advanced Settings

Zone Assignment

By default, all zones are set to Intra-zone FAX Profile 1.

Based on your network diagram, assign the Intra-zone FAX Profiles to the Zone IDs of the zones. If audio compression is required within the same zone, set Intra-Zone Compression to “Yes”. BT GS uses the Inter-zone FAX Profile.

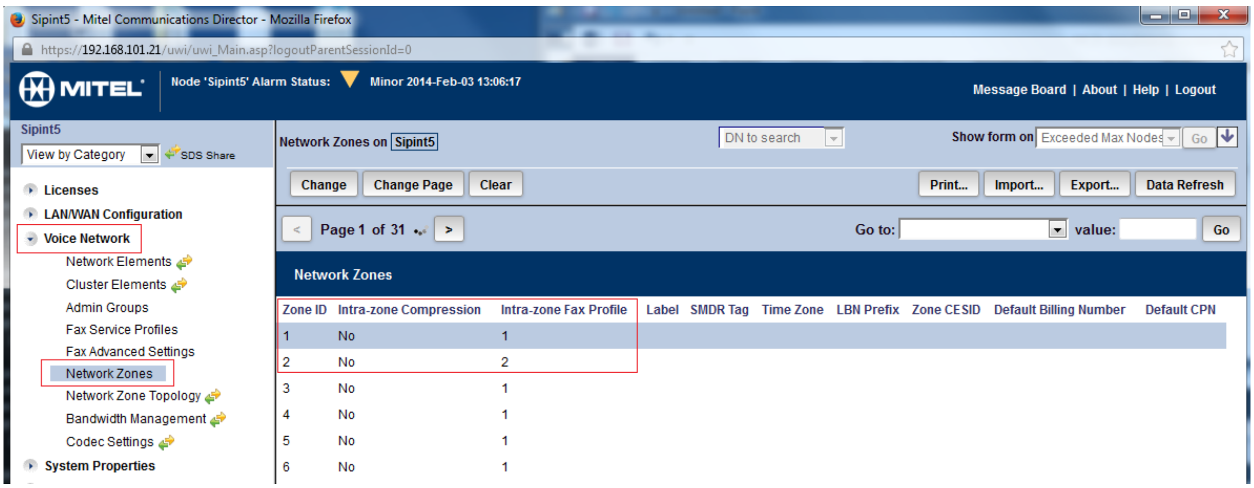


Figure 14 – Zone Assignment

Mitel Border Gateway Setup

MBG Setup

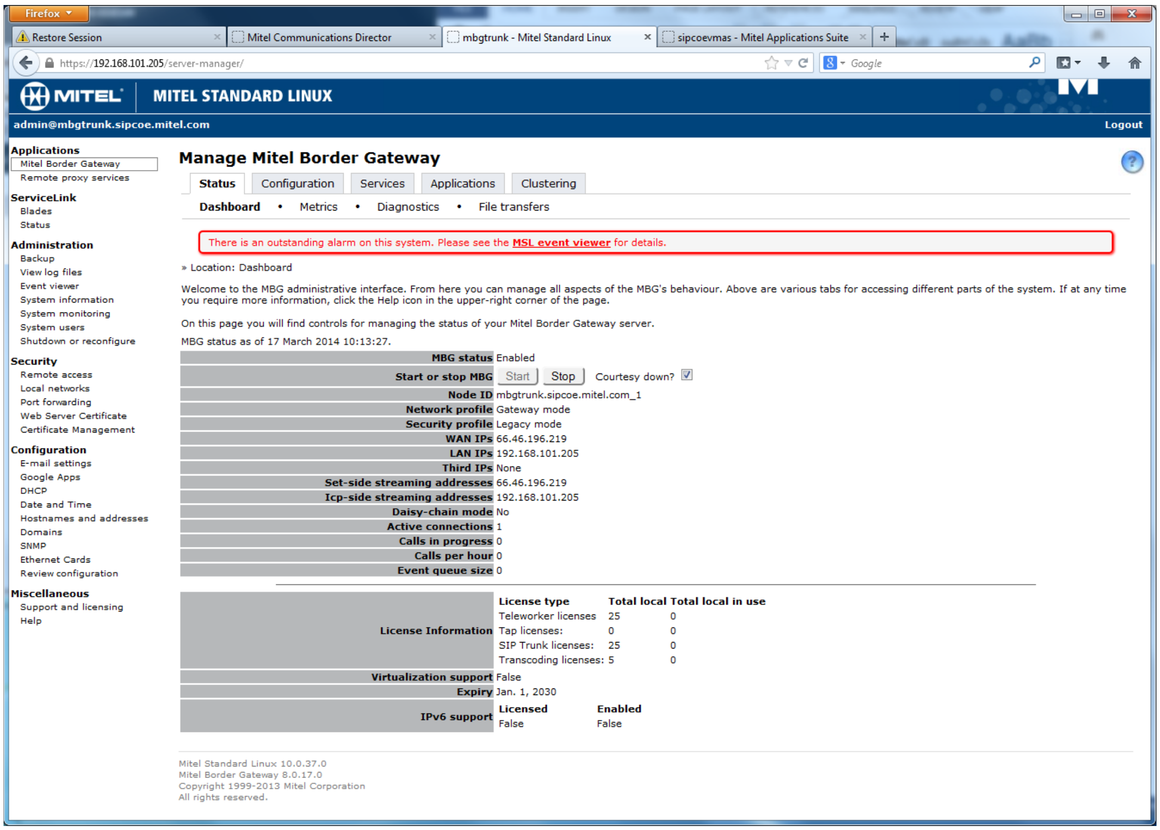


Figure 15 – MBG setup

ICP Setup

To program an MCD into the MBG, click on ICP's→Add an ICP.

Enter a name for the MCD.

Enter the IP address of the MCD and select the Type as MCD.

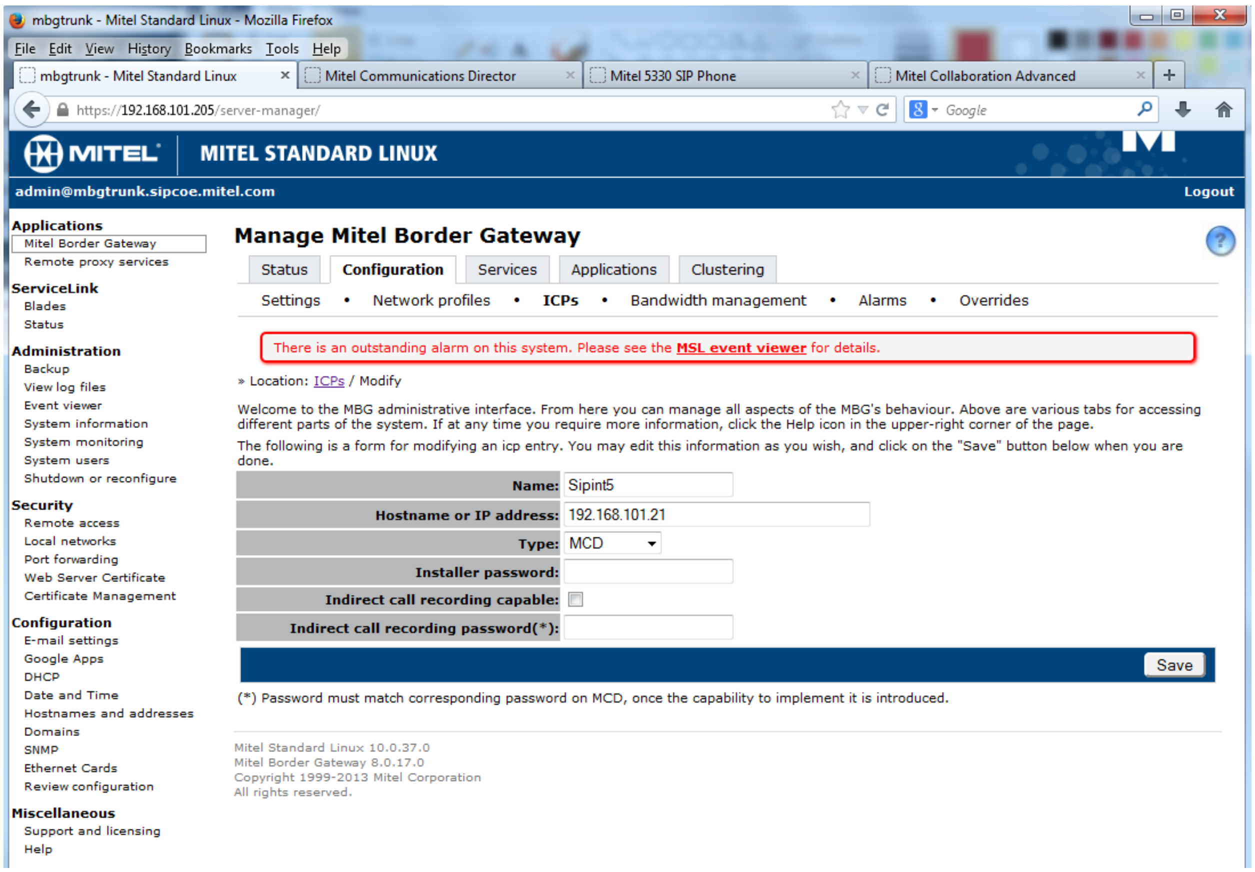


Figure 16 – ICP setup

SIP Trunk Setup

Under the Services tab, click on SIP trunking and then “Add a SIP Trunk”. Enter the SIP trunk’s details as shown in Figure 17:

Name – is the name of the trunk

Remote trunk endpoint address – the public IP address of the provider’s switch or gateway (this address should be given to you by the provider).

Local/Remote RTP framesize (ms) – is the packetization rate you want to set on this trunk

Routing rule one – it allows routing of any digits to the selected Mitel 3300ICP

The rest of the settings are optional and could be configured if required. Click **Save** button

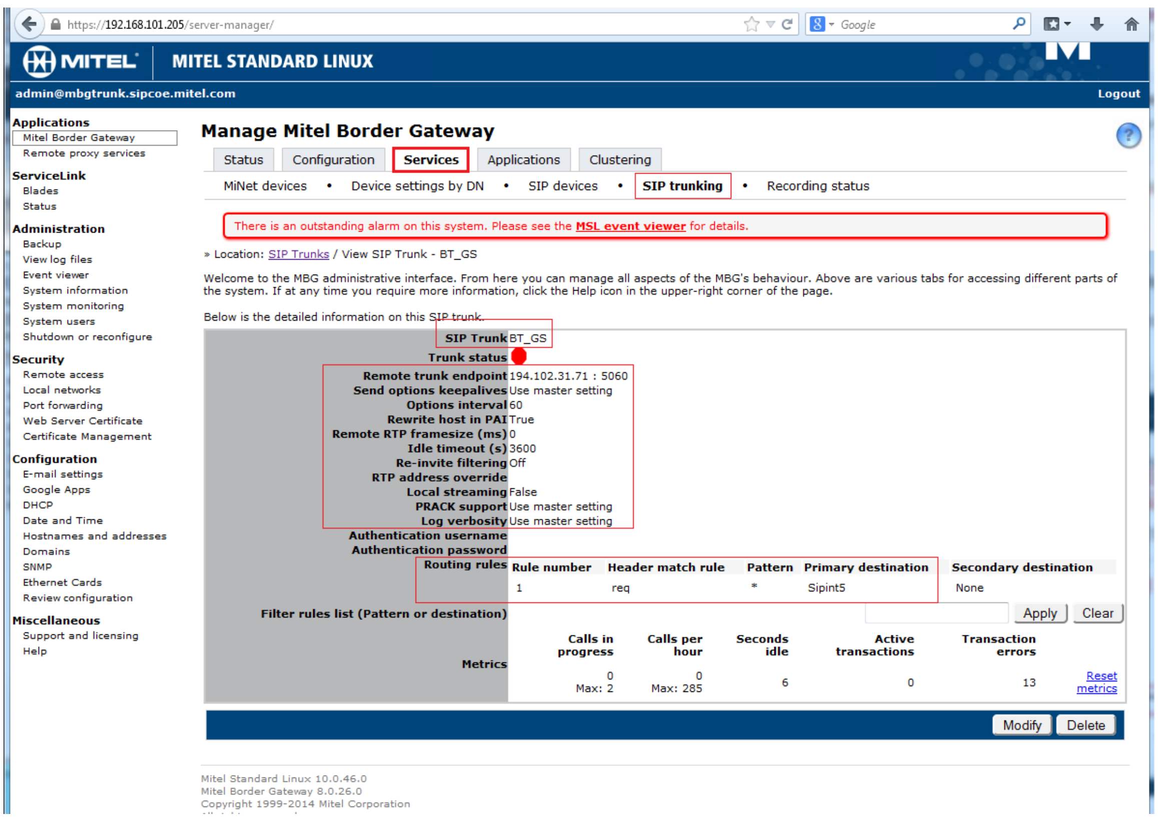


Figure 17 – Services - SIP Trunking setup

Configuration Settings SIP Options

Under the Configurations tab, click on Settings.

SIP options	
SIP support	Enabled
PRACK support	Enabled
Local streaming	False
Send options keepalives	Only behind NAT
Options interval	20
Gap register	True
Set-side registration expiry time	240
ICP-side registration expiry time	900
Challenge methods	invite subscribe
Allowed URI names	cust3-tor.vsac.bell.ca mns.mivid.net 212.23.7.228 194.102.31.71

Figure 18 – Configuration – Settings - SIP Options

Global Headquarters	U.S.	EMEA	CALA	Asia Pacific
Tel: +1(613) 592-2122	Tel: +1(480) 961-9000	Tel: +44(0)1291-430000	Tel: +1(613) 592-2122	Tel: +852 2508 9780
Fax: +1(613) 592-4784	Fax: +1(480) 961-1370	Fax: +44(0)1291-430400	Fax: +1(613) 592-7825	Fax: +852 2508 9232

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